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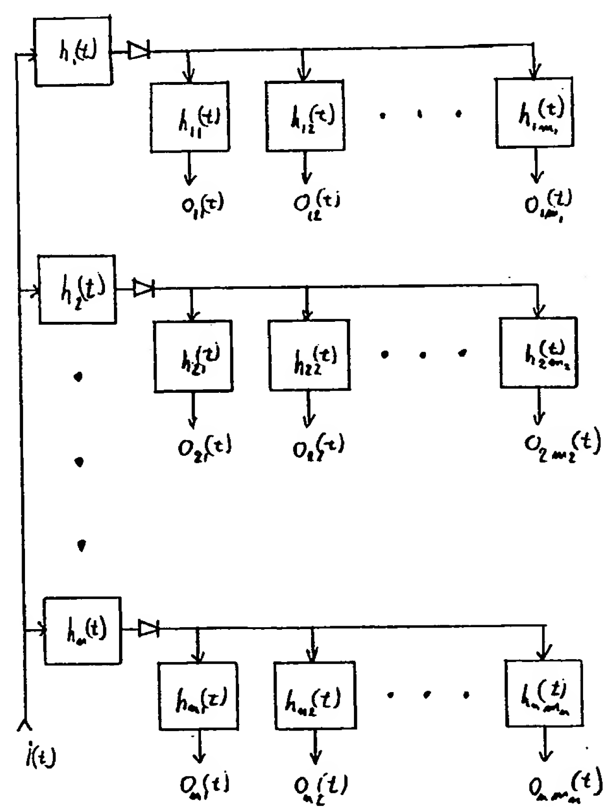
(54) Title: A SIGNAL PROCESSING METHOD TO ANALYSE TRANSIENTS OF SPEECH SIGNALS

(57) Abstract

The present invention is related to a method and an apparatus for determination of a parameter of a system generating a signal containing information about the parameter. The method comprises the step of short time Laplace transforming the signal and may be utilised for classifying the system in question in accordance with one or more determined parameters into one class of a set of predefined classes defined by predetermined ranges of values of the parameters. The invention also relates to the use of a shape of energy changes of a signal for identifying or representing features of the system generating the signal. This use may be applied to recognition of sound features perceivable by e.g. a human ear as representing a distinct sound picture. It has for example been found that the signal information relevant to recognition of speech is present in a transient part of the speech signal.



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A SIGNAL PROCESSING METHOD FOR DETERMINATION OF A PARAMETER OF A SYSTEM GENERATING THE SIGNAL

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The present invention relates to a method for determination of a parameter of a system generating a signal containing information about the parameter.

The method may be used for identification of sound or speech signals, such as in speech recognition, or for quality measurement of audio products or systems, such as loudspeakers, hearing aids, telecommunication systems, or for quality measurement of acoustic conditions. The method of the present invention may also be used in connection with speech compression and decompression in narrow band telecommunication.

The method may also be used in analysis of mechanical vibrations generated by a manufactured device during operation e.g. for detection of malfunction of the device.

The method may further be used in electrobiology for example for analysis of neuroelectrical signals such as analysis of signals from an electroencephalograph, an electromyograph, etc.

BACKGROUND OF THE INVENTION

The three documents

HALIJAK C A et al.: "Simple Consequences of the Finite Time Laplace Transform Analysis of the Periodically Reversed Switched Capacitors", CIRCUITS, SYSTEMS, AND SIGNAL PROCESSING, 1985, USA, vol. 4, no. 4, pages 503-511, XP-002105446, ISSN 0278-081X;

BARRETT T W: "The Cochlea as Laplace Analyzer for Optimum (Elementary) Signals", ACUSTICA, Feb. 1978, WEST GERMANY, vol. 39, no. 3, pages 155-172, XP-002105445, ISSN 0001-7884; and

HARBOR R D et al.: "THE LAPLACE TRANSFORM", ENERGY AND INFORMATION TECHNOLOGIES IN THE SOUTHEAST, Columbia, April 9-12, 1989, vol. 1, 9 April 1989, pages 376-379, XP-000076824, IEEE;

offer relevant background art as regards the Laplace transform.

Prior art methods of signal processing are based on a short time Fourier transform of signals and it is assumed that the signals are steady state signals.

In steady state analysis the signal is assumed stationary in the period the signal is analysed and the steady state spectrum is calculated.

In real life steady state signals do not occur and steady state analysis does not provide sufficient knowledge of phenomena within various scientific and technological fields. Consider for example speech analysis. The human ear has the ability to simultaneously catch fast sound signals, detect sound frequencies with great accuracy and differentiate between sound signals in complicated sound environments. For instance it is possible to understand what a singer is singing in an accompaniment of musical instruments.

It is assumed that the cochlea in the human ear can be regarded as comprising a large number of band-pass filters within the frequency range of the human ear.

The time response f(t) for one band-pass filter due to an excitation can be separated into two components, the transient response, $f_t(t)$, and the steady state response, $f_s(t)$, $f(t)=f_t(t)+f_s(t)$.

Traditional signal processing is based on the steady state response $f_s(t)$, and the transient response $f_t(t)$ is assumed to vanish very fast and to be without importance for the perception, see for example "Principles of Circuit Synthesis", McGraw-Hill 1959, Ernest

S. Kuh and Donald O. Pederson, page 12, lines 9-15, where it is stated that:

"only the forced response is considered while the response due to the initial state of the network is ignored".

Thus, when students are introduced to the world of signal analysis, they learn that the transient response, i.e. the response due to the initial state of the network should be ignored because it vanishes within a very short period of time. Furthermore, it is rather difficult to analyse these transient signals by use of traditional linear methods of analysis.

The ability of the human ear to hear very short sounds and at the same time detect frequencies with great accuracy is in conflict with the traditional filterbased spectrum analysis. The time window (twice the rise time) of a band-pass filter is inversely proportional to the bandwidth, $tw=2/(f_u-f_l)$, where f_l is the lower cut-off frequency and f_u is the upper cut-off frequency.

Thus, if a rise time of 5 ms is required the consequence is that the frequency resolution is no better than 400 Hz.

As the detection of these transients is in conflict with a high frequency resolution, the detecting by the human ear of these transients must take place in an alternative manner. It has not been examined how the human ear is able to detect these signals, but it might be possible that the cochlea, when no sounds are received, is in a position of rest, where the cochlea will be very broad-banded. When a sound signal is received, the cochlea may start to lock itself to the frequency component or components within the signal. Thus, the cochlea may be broad-banded in its starting position, but if one or more stable frequencies are received the cochlea may lock itself to this frequency or these frequencies with a high accuracy.

Today it is known that the nerve pulses launched from the cochlea are synchronized to the frequency of a tone if the frequency is less than about 1.4 kHz. If the frequency is higher than 1.4 kHz the pulses are launched randomly and less than once per cycle of the frequency.

Signal processing based on filter bank spectrum analysis is disclosed in GB 2 213 623, which describes a system for phoneme recognition. This system comprises detecting means for detecting transient parts of a voice signal, where the principal object of the transient detection is the detection of a point where the speech spectrum varies most sharply, namely, a peak point. The detection of the peak points is used for more precise phoneme segmentation. The transient analysis of GB 2213623 is based on a spectrum analysis and the change in the spectrum, which is very much different to the transient analysis of the present invention, which is based on a direct transient detection in the time domain.

SUMMARY OF THE INVENTION

The present invention provides an approach, which is different in principle from all known methods for processing signals. The approach taken and some of the results obtained will be explained by of an example in the context of analysis of speech signals.

Speech is produced by means of short pulses generated by the vocal chords in the case of voiced speech and by friction in the vocal tract in the case of unvoiced speech. The pulses are filtered by the vocal tract that acts as a time-varying filter. The output response will consist of quasi steady state terms and also transient terms. The quasi steady state terms will only be damped slightly in the period before the next pulse is generated. The transient terms will be sufficiently damped in the time period before the next pulse is generated.

The speech signal is often assumed to have only quasi steady state terms in the period or time window of the analysis, typically 20-30 ms.

The placement of formants, the formants being energy bands in the short time power spectrum, are calculated by means of a short time spectrum analysis has previously been assumed decisive for speech intelligibility, together with voiced/unvoiced detection, the pitch and the quasi steady state power.

However, a number of observations, which has been performed within the field of auditory perception research, does not conform to the previous assumptions:

Why is it possible to understand and identify a deep male voice through communication channels that have a higher cut-off frequency than the male pitch.

The only difference between the pronunciation of the letters: e, b, d is in the first 1-3 ms of the voice signal and this information will be lost if the analysis have a time window of 20-30 ms.

How can the absolute placement of these formants be decisive when their placement is quite different for different people, particularly between small children and large males.

Why is distortion dominated by odd order harmonics and caused by cross-over distortion in a class B amplifier much more disturbing than distortion dominated by even order harmonics caused by amplitude distortion in a class A amplifier.

The short time power spectrum will not distinguish frequencies from different sources, and tones generated by other sources than the speech signal will act like false formants.

Why does a signal consisting of three tones with the same frequencies as the formants for a vowel not give the slightest perception of the vowel at all? The signal just sounds like three separate tones.

Why is the ear very sensitive to frequency changes of a signal up till about 1000 Hz, changes of +/- 3 Hz can be detected. For frequencies above 1000 Hz, the sensitivity is much smaller.

The research performed by the present applicant leads to suggest that the ear is tone dominant until about 1.4 - 1.6 kHz and transient dominant above. Tone dominant means that the pulses launched from the hair cells as a response to a tone signal are synchronised to the tone signal. Transient dominant means, in the present context, that the hair cells are activated by changes of the energy with rise and fall times of at most 2 ms typical caused by transient pulses.

Regarding speech signals, it is assumed that the quasi steady state terms are in the tone dominant interval of the ear and that the transient terms are in the transient dominant interval. It is believed that the transient terms are very important for speech intelligibility. The transient terms are seen as transient pulses in the speech signal. The rise time and the shape of leading and lagging edges of the envelope of transient pulses in the terms of a profile of damped frequencies describes the sound picture. The shape of the leading and lagging edges, the dynamic changes, change of amplitude, of the transient pulses, voiced/unvoiced detection and the changes of pitch are decisive for speech recognition.

This approach provides a number of advantages with respect to explaining the earlier mentioned speech perception observations.

A natural explanation as to why it is possible to understand and identify a deep male voice through communication channels that have

a higher cut-off frequency than the male pitch is provided. The pitch can be detected as the period between transient pulses.

The absolute placement of formants is not decisive. The damped frequencies profile of the shape of the transient pulse envelope is dominated by damped difference frequencies of the transient terms.

Distortion caused by cross-over distortion in a class B amplifier generates abrupt energy changes (unwanted transients) which are much more disturbing than distortion caused by amplitude distortion in a class A amplifier which do not generate the same abrupt energy changes.

Robust data- or telecommunication is based on modulation. The envelope of transient pulses is a kind of amplitude modulation, transient or impulse response modulation, and will have the same advantages.

It is unlikely that frequencies from other sources will cause interference patterns with the speech signal that gives energy changes with time constants and shapes in the range that is decisive for speech intelligibility. This means that transient modulation will be robust in noisy environments and communication channels.

The ear is probably very sensitive to changes of a frequency up till about 1000 Hz because the nerve pulses are synchronised to the frequency and the period between the pulses is a measure for the frequency. In the high frequency range, where the pulses are not synchronised to the frequency, only placement of the frequency in the cochlea is a measure for the frequency.

According to the invention it has for example been found that the signal information relevant to recognition of speech is present in a transient part of the speech signal. Thus, the method of the present invention may involve a separation of the transient part of

an auditory signal, a generation of a transient pulse corresponding to the transient part, and analysis of the shape of the pulse. In an auditory signal, the corresponding transient pulse may be repeated with time intervals, and the time interval of these periodic transient pulses is normally also analysed or determined.

In real life, the human ear reacts to energy changes at high frequencies in order to recognise phonemes or sound pictures. But in the present method transient pulses corresponding to the energy changes observed by the ear are extracted at these high frequencies, wherefore the transient pulses preferably are transformed to the low frequency range still maintaining the distinct features of the sound pictures or phonemes. Thus, by using the principles of the invention, it is possible to obtain distinct features within auditory signals by examining the transformed low frequency signals.

The invention relates to the use of the shape of energy changes of a signal for identifying or representing features of the system generating the signal for example in recognition of sound features which can be perceived by an animal ear such as a human ear as representing a distinct sound picture are determined.

The method of the present invention provides an expression for the transient conditions of the auditory signal. The method comprises a band-pass filtration of an auditory signal within the frequency range of the human ear and a detection of a low-pass filtered envelope, which envelope then can be analysed with known methods of signal analysis. The envelope is an expression of the transient part of the signal.

The method of signal analysis, which should be used when analysing the envelope, and the characteristics of the band-pass filter, which should be selected, will depend on the purpose of the analysis. The purpose may be speech recognition, quality-

measurement of audio products or acoustic conditions, and narrow band telecommunication.

The invention also relates to a system for processing a signal to reduce the bandwidth of the signal with substantial retention of the information of the signal. The system may further comprise means for extracting the transient component of the auditory signal, and it may comprise means for detecting an envelope of the transient component.

A signal may be separated into a sum of impulse responses generated by poles and zeroes in the system that has generated the signal, if the time between the excitation pulses are sufficient long compared to the duration of the impulse responses for the system.

In WO 94/25958 it is shown that the envelope of the transient component in a speech signal is very important for its recognition and it is shown that the envelope of the impulse response will contain exponential functions and difference frequencies defined by the impulse response.

A method based on damped sinus functions to extract important features from the envelope signal is described, and examples where the method is used on speech signals shows that the features are important in speech analysis.

Before entering into a more detailed explanation of features of the method of the invention, a few definitions will be given:

In short time analysis the transient component in a signal is a matter of definition. For auditory signals, the idea is to obtain an expression that gives a response corresponding to the response in the cochlea to an abrupt change in the signal energy. An abrupt change in the signal energy corresponds to the transient component in the auditory signal. Thus, in the present context, the term "transient component" designates any signal corresponding to an

abrupt energy change in an auditory signal. The transient component holds the signal information to be analysed and in order to analyse this information the transient component may be transformed to a corresponding transient pulse having a distinct shape. Thus, in the present context, the term "transient pulse" refers to a pulse having a distinct shape and substantially holding the information of the transient component of the auditory signal and thus corresponding to an abrupt change in the energy of the auditory signal. As mentioned above the transient part of a sound signal may be repeated with time intervals and thus, in the present context, the term "periodic" when used in combination with a transient component, response or pulse designates any transient component, response or pulse being repeated with intervals.

The term "shape" designates any arbitrary time-varying function (which is time-limited or not time-limited) and which, within a given time interval T_p has a distinctly different amplitude level in comparison with the amplitude level outside the interval. Thus, T_p is the duration of the shape function when the shape function is time-limited, or the duration of the part of the function which has a distinctly different amplitude level in comparison with the amplitude level outside the time interval.

In order to extract information from the shape of the energy changes, one broad aspect of the invention relates to represent the shape of the energy changes by the short time Laplace transform of a transient pulse of the signal. However, several methods can be applied in order to obtain a transient pulse corresponding to the change in energy, but it is preferred that an envelope detection is being used, where the envelope preferably should be detected from a transient response of the energy change in the auditory signal.

The energy change representing the distinct sound picture can be a phoneme or vowel or any other sound which gives a sudden energy change in an auditory signal.

It is also an aspect of the invention to provide a method for identifying, in an auditory signal, energy changes which can be perceived by an animal ear such as a human ear as representing a distinct sound picture, the method comprising comparing the shape of energy changes of the signal with predetermined energy change shapes representing distinct sound pictures. For the identification it is preferred that the shape of the energy changes are represented by the shape of a transient pulse of the signal, and it is furthermore preferred that the shape of the transient pulse should be obtained by an envelope detection of a transient response of the energy change in the auditory signal.

The invention also relates to a method for processing a signal so as to reduce the bandwidth of the signal with substantial retention of the information of the signal, comprising extracting a transient part of the signal. The method may further comprise detecting an envelope of the transient part of the signal.

Known methods of processing signals are based on a short time Fourier transform of signals, and it is assumed that the signals are steady state signals.

In steady state analysis the signal is assumed stable in the period the signal is analysed, and the steady state spectrum is calculated.

In WO 94/25958 it is disclosed that transient pulses are important for speech coding and decoding in narrow band communication, for speech recognition and synthesis, and for sound quality in auditory products (i.e. loudspeakers, amplifiers and hearing aids).

An important part of a transient signal is the exponential functions or damping ratios or time constants. The damping ratio is the reason that the impulse response has a finite duration. The fact that the transient signal is important for auditory perception indicates that the response from the hair cells is dependent on the

time constants. If this is the case, it is possible that the damping ratios in the response from nerve cells in general are important for the human nerve system.

Transient signals are also important in many other applications, among others signals generated by impacts from defects in rolling bearings and gearboxes.

Based on the transient signal, it is possible to determine the natural time constants and frequencies in the system generating the signal. Further it is possible to determine the excitation pulses of the system.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 shows a time-domain representation of a linear time-invariant system,

Fig. 2 shows the impulse response of a Butterworth low-pass filter of 3. order and a cut-off frequency at 700 Hz,

Fig. 3 shows the response with the filter relaxed for $t < 0$ and with a 4000 Hz tone as input at $t \geq 0$,

Fig. 4 shows the s-plane with poles and the zero for $H(\sigma,\omega)$,

Fig. 5 shows $H(\sigma,\omega)$ for ω_1 and ω_2 analysed parallel with the σ axis,

Fig. 6 shows transient characteristics in speech signals,

Figs. 7-12 show processed speech signals,

Fig. 13 shows a schematic of a filter bank according to the present invention.

DETAILED DESCRIPTION OF THE DRAWINGS

The importance of the transient part of a signal has been an overlooked phenomenon in signal analysis.

The response of a linear system to either an impulse or a step function is defined by its transient response properties.

The relationship between the input and the output for the linear time-invariant system shown in Fig. 1 can be written as the convolution of the input signal and the impulse response of the system:

$$v_o(t) = \int_{-\infty}^{t} v_i(x)h(t-x)dx \qquad (1)$$

If the system is initially relaxed and the input signal $v_i(t)$ is zero for $t < 0$ then the lower integration limit of Eq. (1) can be replaced with zero. Eq. (1) then shows the important role played by the impulse response in terms of the actual signal processing that is performed by the system. It states that the input signal is weighted or multiplied by the impulse response at every instant in time and, at any specific point in time, the output is the summation or integral of all past weighted inputs.

The impulse response of a real system has a finite duration and the transient response has the same duration. Fig. 2 shows the impulse response of a Butterworth low-pass filter of 3. order and a cut-off frequency at 700 Hz. Fig. 3 shows the response with the filter relaxed for $t < 0$ and with a 4000 Hz tone as input at $t \geq 0$.

In many processes $v_i(t)$ will be a pulse with a short duration and $v_i(t) \approx 0$ before the next pulse will be generated.

The Laplace transform of a signal $v(t)$ is defined by

$$L(s) = \int_0^\infty v(t)\mathrm{e}^{-st}dt \qquad (2)$$

$$= \int_0^\infty v(t)e^{-(\sigma+j\omega)t}dt$$

If $v(t)$ is the impulse response $h(t)$ for a system with 2 complex poles

$$h(t) = e^{-(\sigma_0+j\omega_0)t} + e^{-(\sigma_0-j\omega_0)t}, \quad t > 0 \qquad (3)$$

and 0 for $t < 0$ and $s \neq -(\sigma_0 \pm j\omega_0)$.

The Laplace transform is

$$H(s) = \frac{s+\sigma_0}{(s+\sigma_0+j\omega_0)(s+\sigma_0-j\omega_0)}$$

or

$$H(\sigma,\omega) = \frac{\sigma+\sigma_0+j\omega_0}{(\sigma+\sigma_0+j(\omega+\omega_0))(\sigma+\sigma_0+j(\omega-\omega_0))} \qquad (4)$$

From Eq.(4) it is seen that for $(\sigma,\omega) \to (-\sigma_0, \pm\omega_0)$, $H(\sigma,\omega) \to \pm\infty$.

This is a well-known phenomenon and a logical consequence of this is as follows:

If the signal analysed is dominated by the impulse response of the system generating the signal, it is possible to determine the natural time constants and frequencies for the system.

Fig. 5 shows a plot of $H(\sigma,\omega)$ for $\omega = \omega_1$ and $\omega = \omega_2$.

Analysing a signal along or parallel with the jω axis will give a frequency profile for a given σ.

Analysing a signal along or parallel with the σ axis will give a time constant profile for a given jω.

If a signal has a time constant profile with significant variations for specific frequencies, the signal is transient dominated. Opposite if the signal does not vary significantly for any frequency, the signal is steady state dominated.

A short time Laplace transform is defined by:

$$L(\sigma,\omega,t)=\int_0^t v_i(t-\lambda)e^{-(\sigma+j\omega)\lambda}d\lambda \qquad (5)$$

in which v_i is the signal, L is the transformed signal, σ is a time constant, and ω is an angular frequency.

It is not possible to calculate the short time Laplace transform in the same way as DFT in the discrete time domain because two arbitrary exponential functions, e^{at} and e^{bt}, are not orthogonal with respect to each other.

The short time Fourier analysis in the analogue time domain is based on a filter bank method. In this paper an equivalent method will be developed for the Laplace transform.

From Eq.(1) and Eq.(3):

$$v_o(t)=\int_0^t v_i(t-\lambda)e^{-(\sigma+j\omega)\lambda}d\lambda$$

$$+\int_0^t v_i(t-\lambda)e^{-(\sigma-j\omega)\lambda}d\lambda \qquad (6)$$

$$v_o(t) = V(\sigma,\omega,t) + V^*(\sigma,\omega,t) = u(t) + u^*(t)$$

where $u^*(t)$ is the complex conjugate of $u(t)$ and we have

$$\mathrm{Re}\left[L(\sigma,\omega,t)\right] = \tfrac{1}{2}v_o(t) \qquad (7)$$

From Eq.(6) and Eq.(7) it is seen that filtering the signal $v_i(t)$ by a filter with the impulse response $h(\sigma,\omega,t)$ with 2 complex poles will represent the reel part of the short time $L(\sigma,\omega,t)$ transform.

If we let $v_i(t)$ be equal to the impulse response of a single pole we have

$$\begin{aligned} u(t) &= \int_0^t ke^{-(\sigma_0+j\omega_0)(t-\lambda)}e^{-(\sigma+j\omega)\lambda}d\lambda \\ &= ke^{-(\sigma_0+j\omega_0)t}\int_0^t e^{(\sigma_0+j\omega_0)\lambda}e^{-(\sigma+j\omega)\lambda}d\lambda \qquad (8) \\ &= \frac{k(e^{-(\sigma+j\omega)t} - e^{-(\sigma_0+j\omega_0)t})}{(\sigma-\sigma_0)+j(\omega-\omega_0)} \end{aligned}$$

and from Eq.(7) we have

$$\begin{aligned} v_o(t) = &-\frac{2k(\sigma-\sigma_0)(e^{-\sigma t}\cos(\omega t) - e^{-\sigma_0 t}\cos(\omega_0 t))}{(\sigma-\sigma_0)^2+(\omega-\omega_0)^2} \\ &+\frac{2k(\omega-\omega_0)(e^{-\sigma t}\sin(\omega t) - e^{-\sigma_0 t}\sin(\omega_0 t))}{(\sigma-\sigma_0)^2+(\omega-\omega_0)^2} \qquad (9a) \end{aligned}$$

or

$$\frac{v_o(t)}{2k} = \frac{e^{-\sigma_0 t}((\sigma-\sigma_0)\cos(\omega_0 t) - (\omega-\omega_0)\sin(\omega_0 t))}{(\sigma-\sigma_0)^2+(\omega-\omega_0)^2}$$

$$\frac{-e^{-\sigma t}((\sigma-\sigma_0)\cos(\omega t)-(\omega-\omega_0)\sin(\omega t))}{(\sigma-\sigma_0)^2+(\omega-\omega_0)^2} \qquad (9b)$$

Eq.(9) is not defined for $(\sigma,\omega)=(\sigma_0,\omega_0)$ but from (8) we have in this case

$$u(t)=ke^{-(\sigma_0+j\omega_0)t}\int_0^t d\lambda$$

$$=kte^{-(\sigma_0+j\omega_0)t}$$

and

$$v_o(t)=2kte^{-\sigma_0 t}\cos(\omega_0 t) \qquad (10)$$

and we have $v_o(t)\to 0$ for $t\to\infty$.

Eq.(9) shows that the gain is inversely related to $\sigma-\sigma_0$ and $\omega-\omega_0$, and when (σ_0,ω_0) is far from (σ,ω) and $e^{-\sigma t}-e^{-\sigma_0 t}$ is small, $v_o(t)\approx 0$. For $(\sigma_0,\omega_0)\leftarrow(\sigma,\omega)$ $v_o(t)$ will have Eq.(10) as the limit. It is not immediately to see if Eq.(9) has the maximum energy for $(\sigma_0,\omega_0)\leftarrow(\sigma,\omega)$.

In the DC domain Eq.(9) can be written as

$$v_o(t)=2k\frac{(e^{-\sigma_0 t}-e^{-\sigma t})}{\sigma-\sigma_0} \qquad (11)$$

The maximum for $v_o(t)$ can be found as follows

$$\frac{dv_o}{dt}=\frac{1}{\sigma-\sigma_0}\left[\sigma e^{-\sigma t}-\sigma_0 e^{-\sigma_0 t}\right]=0$$

when

$$t_m = \frac{\log(\sigma) - \log(\sigma_0)}{\sigma - \sigma_0} \qquad (12)$$

and Eq.(11) will have the maximum for this value.

It can be shown that $t_m \to \frac{1}{\sigma_0}$ when $\sigma \to \sigma_0$.

When $\sigma \approx \sigma_0$ we will have the approximated maximum with $t = \frac{1}{\sigma_0}$

$$v_o(\tfrac{1}{\sigma_0}) = 2k\frac{(e^{-1} - e^{-\frac{\sigma}{\sigma_0}})}{\sigma - \sigma_0} \qquad (13)$$

From Eq.(13) it can be shown that

$$v_o \to \frac{2ke^{-1}}{\sigma_0} \text{ for } \sigma \to \sigma_0$$

In Eq.(11) $e^{-\sigma_0 t}$ represent the signal to be analysed and $e^{-\sigma t}$ the filter. Table 1 shows the result with a filter having σ = 100 s^{-1} and the signal varying from 1 to 10000 s^{-1}

It is not surprising that the convolution acts as a low-pass filter. The important fact is that the exponential function in the DC domain in some way acts as frequencies do in the frequency domain.

In table 1 $v_{o1}(t_m)$ is the result of a convolution where the signal is differentiated. The result is, as expected, a high-pass filter.

If we look on Eq.(9a) without exponential functions it can be written as

$$v_0(t) = \frac{2k(\sin(\omega t) - \sin(\omega_0 t))}{\omega - \omega_0} \tag{14}$$

it is seen that for $\omega \to \infty$ we will have $v_o \to 0$.

$\sigma : 100\ s^{-1}$			
σ_0	t_m	$v_o(t_m)$	$v_{o1}(t_m)$
s^{-1}	s		
1	0,046516871	0,954548457	0,009545485
10	0,025584279	0,774263683	0,077426368
100	0,010000000	0,367879441	0,367879441
1000	0,002558428	0,077426368	0,774263683
10000	0,000465169	0,009545485	0,954548457

Table 1 $v_o(t_m)$ is given by Eq.(11, 12) and normalised by σ and 2k. $v_{o1}(t_m)$ is a convolution where the signal is differentiated and normalised by 2k.

For $\omega << \omega_0$ we will have

$$\dot{v}_o \cong \frac{2k(\sin(\omega t) - \sin(\omega_0 t))}{\omega_0} \tag{15}$$

It can be shown that for $\omega \to \omega_0$ we will have

$$v_o(t) \to 2kt\cos(\omega_0 t) \tag{16}$$

This result is as expected unstable.

In transient analysis only the beginning of the signal is of interest, and if $\omega_0 >> 1$ Eq.(14) will act as a band-pass filter.

Speech processing is based on fast energy pulse generated by the vocal cords or by friction in the articulation channel weighted by

the impulse response in the articulation channel. The rise time for the excitation pulses has to be sufficient faster than the rise time of the energy of the impulse response.

The shape of energy pulses are important features in speech. If the time between the pulses is periodical it is voiced speech, and if not it is unvoiced speech. For some phonemes abrupt changes in the energy pulses are important.

From WO 94/25958 it is known that the shape of the energy pulses are important for speech recognition, especially the leading edge. In the following a method to extract features will be developed based on an envelope detection.

The convolution expressed in Eq.(9) can be regarded as a response from 2 poles in the articulation channel excited by an impulse. If $\sigma_0 \approx \sigma$ we have from Eq.(9a)

$$v_o(t) \cong \frac{e^{-\sigma t}}{(\omega - \omega_0)}\left(\sin(\omega t) - \sin(\omega_0 t)\right) \qquad (17)$$

The envelope is defined as

$$e(t) = \sqrt{u^2(t) + \bar{u}^2(t)}$$

where

$$\bar{u}(t) = u(t) * \frac{1}{\pi t}$$

is the Hilbert Transform.

The envelope of Eq.(17) is then

$$e_o(t) = \frac{e^{-\sigma t}}{|\omega - \omega_0|}\sqrt{(\sin(\omega t) - \sin(\omega_0 t))^2 + (-\cos(\omega t) + \cos(\omega_0 t))^2}$$

$$= \frac{e^{-\sigma t}}{|\omega - \omega_0|} \sqrt{2(1 - \cos(\omega - \omega_0)t)}$$

$$\cong \frac{\sqrt{2}e^{-\sigma t}}{|\omega - \omega_0|} \left(1 - \tfrac{1}{2}\cos((\omega - \omega_0)t)\right) \qquad (18)$$

The approximation is acceptable because $|\cos((\omega - \omega_0)t)| \leq 1$

As expected the envelope has a component with the difference frequency of the 2 frequencies.

The conclusion is that we can expect to find damped difference frequencies in the envelope of the transient component.

To detect the damped difference frequencies a filter bank is used. The features might be detected as a convolution between the transient pulse and the impulse response of the filters.

In general form the impulse response can be written as

$$h(t) = ke^{-\lambda t} \sin(f(\lambda)t + \phi)$$

Where $\sigma = \lambda$ and $\omega = f(\lambda)$.

In the following analysis $f(\lambda) = 1.5\lambda$, $k = \omega = 1.5\lambda$, and $\phi = 0$ are selected and we have

$$h(t) = 1.5\lambda e^{-\lambda t} \sin(1.5\lambda t) \qquad (19)$$

By selecting $\omega = 1.5\sigma$ Eq.(19) will act as a band-pass filter with a low Q in relation to the frequencies. Other ratios ω/σ than 1.5 may be selected and it is presently preferred that the ratio (ω/σ) ranges from 0.5 to 2.5. The exponential function gives the advance

that it acts like natural time window that ensure that the signal is natural damped. The value of the parameters are selected by studying rise times in important transient pulses and by experiments.

Fig. 6 shows transient characteristics in speech signals. The top figure shows 50 ms of an "a" in "hard key" pronounced by a female.

The second signal is a band-pass filtration of the speech signal. The band-pass filter is a Butterworth filter with 6 poles and a band width from 2150 to 3550 Hz. This frequency band contains important transient pulses in the sensitive frequency interval of the ear.

The third signal is a energy detection of the transient characteristics of the band-pass filtered speech signal. The detection is an envelope detection performed by means of a rectification and a low-pass filtration of the signal. The filter is a Butterworth filter with 3 poles and a cut-off frequency at 700 Hz.

In WO 97/09712 a method for automatically detecting the leading edges is disclosed. The method uses the maximum slope of the leading edge as reference, and the point before the maximum slope where the slope is less than a given threshold (10-20 % of the maximum slope) the leading edge is defined to begin.

The transient (envelope) signal in Fig.(6) has a DC component, which does not contain any information. Therefore it is preferred that the signal is differentiated before it is analysed e.g. by the filter bank shown in Fig. 13.

In Fig. 13, the filters ($h_1(t)$, $h_2(t)$,…, $h_n(t)$) in the filter bank connected between the input and the envelope detectors are band-pass filters having bandwidths corresponding to the bandwidths of

the band-pass filters of the cochlea and having centre frequencies ranging from 1400 Hz to 6500 Hz.

The output signals $o_{ij}(p)$ from the filter bank shown in Fig. 13 is calculated by:

$$h_{ij}(p) = 1.5\lambda_m e^{-\lambda_m p}\sin(\lambda_m p)\,, \qquad i=0,1,\ldots,N-1 \quad j=0,1,\ldots,M-1$$

$$h_{ij}(p) = 0\,, \qquad p < 0$$

$$o_{ij}(p) = \sum_{k=0}^{P-1} t'(k) h_m(p-k)\,, \qquad p=0,1,\ldots,P-1$$

$m=0,1,\ldots,M-1$ and M is the number of band-pass filters with a low Q in the filter bank connected between the outputs and the envelope detectors, $p = 0,1,\ldots,P-1$ is the sample number, t' is the differentiated transient signal, and λ_m is the filter bank parameter and it is normalised by the sampling frequency.

In the analysis M is selected to 10 and $1500 \le \lambda'_m \le 12000$ s^{-1}, λ'_m is not normalised. By this we have $1885 \le \omega_m \le 18850$ s^{-1} or $300 \le f_m \le 3000$ Hz.

This filtering process is not done in the cochlea but in the hair cells or in the nerve system behind the hair cells.

The Figs. 7, 8, 9, 10, 11, and 12 show the output of the processing of transient signals in the vowels "a", "o", "i" in "hard key" and "soft key" pronounced by a female and a male. Further the figures show plots of maxima of the output signals as a function of the time constant σ of the corresponding filter.

The figures show that maximum curves are very much alike for the same vowels, independent of whether a female or male pronounces it.

With a library of templates and a distance measure it is possible to identify the sound picture, and it can be used for speech recognition and narrow band communication.

Thus, according to the invention a method and an apparatus are provided for determination of a parameter of a system generating a signal containing information about the parameter, in which the signal is short time transformed substantially in accordance with

$$L(\sigma,\omega,t)=\int_0^t v_i(t-\lambda)e^{-(\sigma+j\omega)\lambda+\varphi}d\lambda$$

in which v_i is the signal, L is the transformed signal, σ is a time constant, ω is an angular frequency, and φ is a phase, or, in accordance with another transformation which will give rise to an $L'(\sigma,\omega,t)$ which in time intervals within which $L(\sigma,\omega,t)$ is larger than 10% of its maximum value is not more than 50% different from the result given by the short time Laplace transformation.

In narrow band communication the transient pulses have to be identified and coded, and the decoder will contain a library of filters with corresponding transient responses. The decoder library could also contain the transient responses.

The present invention also relates to measurement of mechanical vibrations e.g. when testing devices that generate mechanical energy during operation, such as mechanical devices with moving parts, such as compressors for refrigerators, electric motors, household machines, electric razors, combustion engines, etc, etc.

For example, it is known that measurement of vibration generated or sound emitted by a device during operation can be useful for detection of malfunction of the device. Certain failures may

generate sound or vibration of specific characteristics that can be recognised.

The method may also comprise steps of classification for classifying a tested device in accordance with the determined parameters into one class of a set of predefined classes. Each predefined class may be defined by a set of upper and lower limits for specific parameters determined according to the method. A device may then be classified as belonging to a certain class if its corresponding parameter values lie within corresponding upper and lower limits of the class.

Each class may correspond to a specific type of failure of the device. For example, shaft imbalance, wheel imbalance, crookedness, imperfections of teeth in cogs, tight bearing, loose bearings, etc, may cause the device to vibrate in different characteristic ways, whereby a characteristic mechanical vibration or sound is generated for each type of failure. The type of failure of the device may then be detected by comparing determined device parameters with corresponding parameter values of various predetermined classes.

The upper and lower limits of a specific class of devices may be determined by testing a set of devices known to belong to that class. For example, the upper limits may be determined as the average of specific parameter values plus three times the standard deviation. Likewise, the lower limits may be determined as the average of parameter values minus three times the standard deviation.

CLAIMS

1. A method for determination of a parameter of a system generating a signal containing information about the parameter, comprising the step of short time transforming the signal substantially in accordance with

$$L(\sigma,\omega,t)=\int_0^t v_i(t-\lambda)e^{-(\sigma+j\omega)\lambda+\varphi}d\lambda$$

in which v_i is the signal, L is the transformed signal, σ is a time constant, ω is an angular frequency, and φ is a phase.

2. A method according to claim 1, wherein the step of transforming comprises filtering the signal v_i with a filter having a pole at $\sigma + j\omega t$ and a pole at $\sigma - j\omega t$.

3. A method according to claim 1 or 2, comprising steps of transforming the signal v_i for a plurality of sets of σ and ω values.

4. A method according to any of the preceding claims, further comprising the step of determining a maximum of at least one transformed signal $L(\sigma,\omega,t)$.

5. A method according to any of the preceding claims, further comprising the step of comparing transformed signals L with corresponding reference signals in order to determine parameters of the system.

6. A method according to any of the preceding claims, further comprising a step of pre-processing the signal before the step of short time transforming, the pre-processing being selected from the

group consisting of filtering, rectification, differentiation, integration, and amplification.

7. A method of transmitting a signal containing information of a set of parameters of a system generating the signal, comprising processing the signal according to any of the preceding claims and further comprising the step of transmitting the determined parameter values.

8. A method according to claim 7 further comprising the step of generating a copy of the signal from the transmitted parameter values.

9. A method of transmitting a signal containing information of a set of parameters of a system generating the signal, comprising processing the signal according to any of the preceding claims and further comprising the steps of

comparing the signal with a library of signals generated for a predetermined set of parameter values by the system,

selecting the library function that constitutes the best match to the signal, and

transmitting an identification signal that identifies the matching library function.

10. A method according to claim 9, further comprising the steps of receiving the identification signal and generating the corresponding library signal.

11. A method of classifying a system according to one or more parameters of the system generating a signal containing information about the one or more parameters, comprising determining the one or more parameters according to any of claims 1-6 and further comprising the step of classifying the system in accordance with

the one or more determined parameters into one class of a set of predefined classes defined by predetermined ranges of values of the parameters.

12. A method for communicating an auditory signal, comprising processing the signal by the method according to any of claims 1-6, transmitting the processed signal, and receiving the processed signal by a receiver.

13. A method according to claim 12, wherein, prior to transmission of the processed signal, the signal is coded into a digital representation, and the coded signal is decoded in the receiver so as to reestablish transient pulse shapes perceived by an animal ear such as a human ear as representing the distinct sound pictures of the auditory signal.

14. A method according to claim 13, wherein the digital transmission is performed at a bandwidth of at the most 4000 bits per second.

15. A method according to claim 14, wherein the bandwidth is at the most 2000 bits per second.

16. A method according to claim 15, wherein the bandwidth is in the interval of 800-2000 bits per second.

17. A method according to any of claims 13-16, wherein a second and further pulses in a sequence of identical pulses are represented by a digital value indicating repetition.

18. A method according to any of claims 1-6, comprising filtering the signal v_i in a filter bank comprising a plurality of band-pass filters interconnected in parallel with centre frequencies ranging from 1400 Hz to 6500 Hz, each of which is connected in series with an envelope detector and a filter bank comprising a plurality of low-pass filters interconnected in parallel and having cut-off

frequencies ranging from 300 Hz to 3000 Hz and time constants σ ranging from 1500 s^{-1} to 12000 s^{-1}.

19. An apparatus for determination of a parameter of a system generating a signal containing information about the parameter, comprising a processor that is adapted to short time transform the signal substantially in accordance with

$$L(\sigma,\omega,t)=\int_0^t v_i(t-\lambda)e^{-(\sigma+j\omega)\lambda+\varphi}d\lambda$$

in which v_i is the signal, L is the transformed signal, σ is a time constant, ω is an angular frequency, and φ is a phase.

20. An apparatus according to claim 19, wherein the processor comprises a filter for filtering the signal v_i and having a pole at $\sigma + j\omega t$ and a pole at $\sigma - j\omega t$.

21. An apparatus according to claim 19 or 20, wherein the processor comprises a plurality of filters for filtering the signal v_i, each filter having a different set of σ and ω values.

22. An apparatus according to claim 19, wherein the apparatus comprises a communication channel transmitter, and the processor is adapted to determine the one or several parameters of the system, and

to transmit the one or several system parameters over a wireless or a cable communication channel.

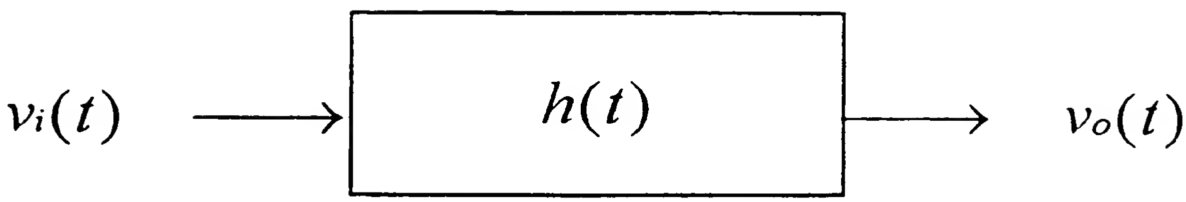

Fig. 1

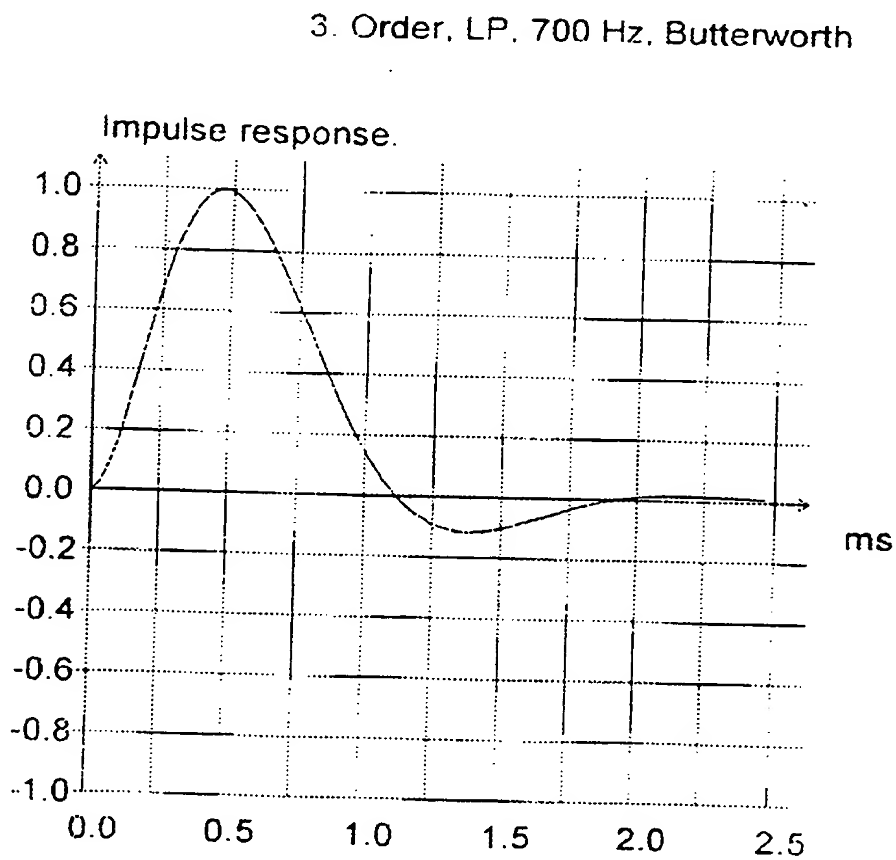
3. Order. LP. 700 Hz. Butterworth
Impulse response.
1.0
0.8
0.6
0.4
0.2
0.0
-0.2
-0.4
-0.6
-0.8
-1.0
0.0
0.5
1.0
1.5
2.0
2.5
ms

Fig. 2

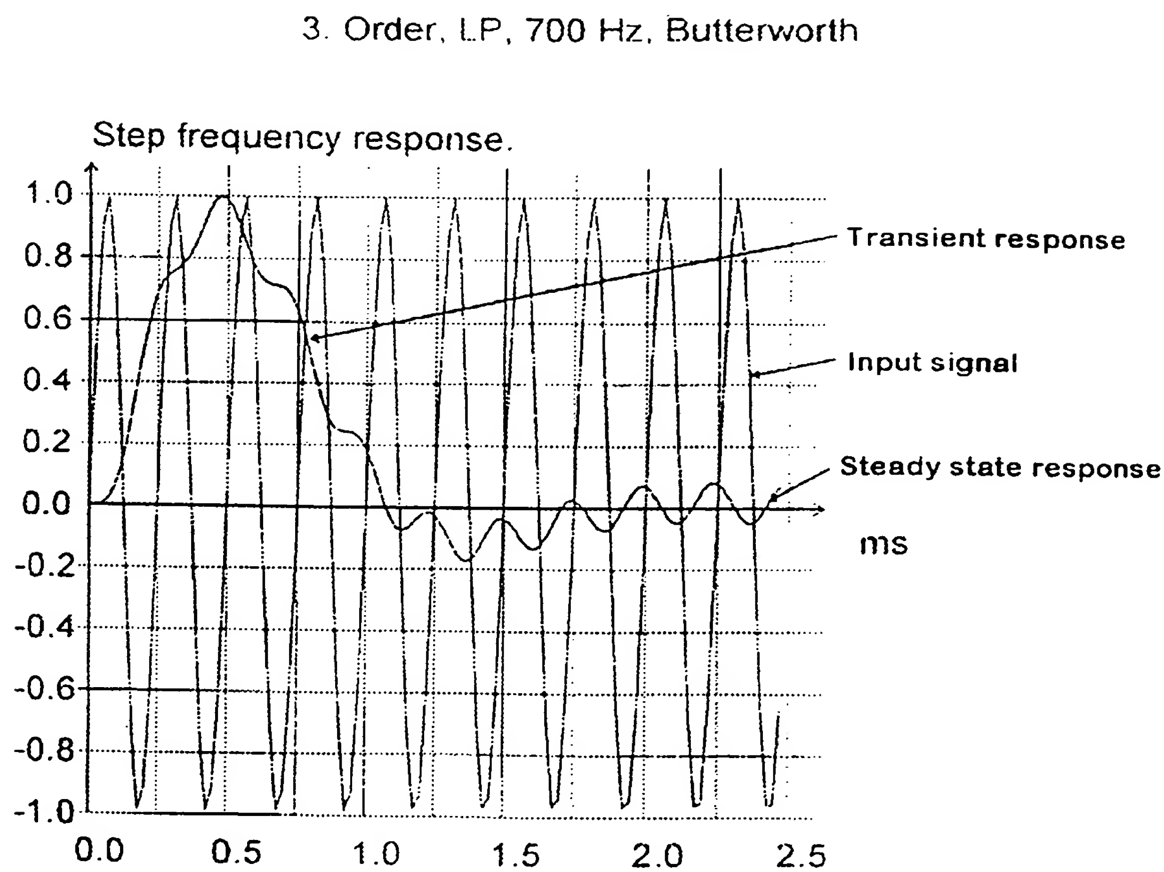
3. Order, LP, 700 Hz, Butterworth
Step frequency response.
1.0
0.8
0.6
0.4
0.2
0.0
-0.2
-0.4
-0.6
-0.8
-1.0
0.0
0.5
1.0
1.5
2.0
2.5
ms
Transient response
Input signal
Steady state response

Fig. 3

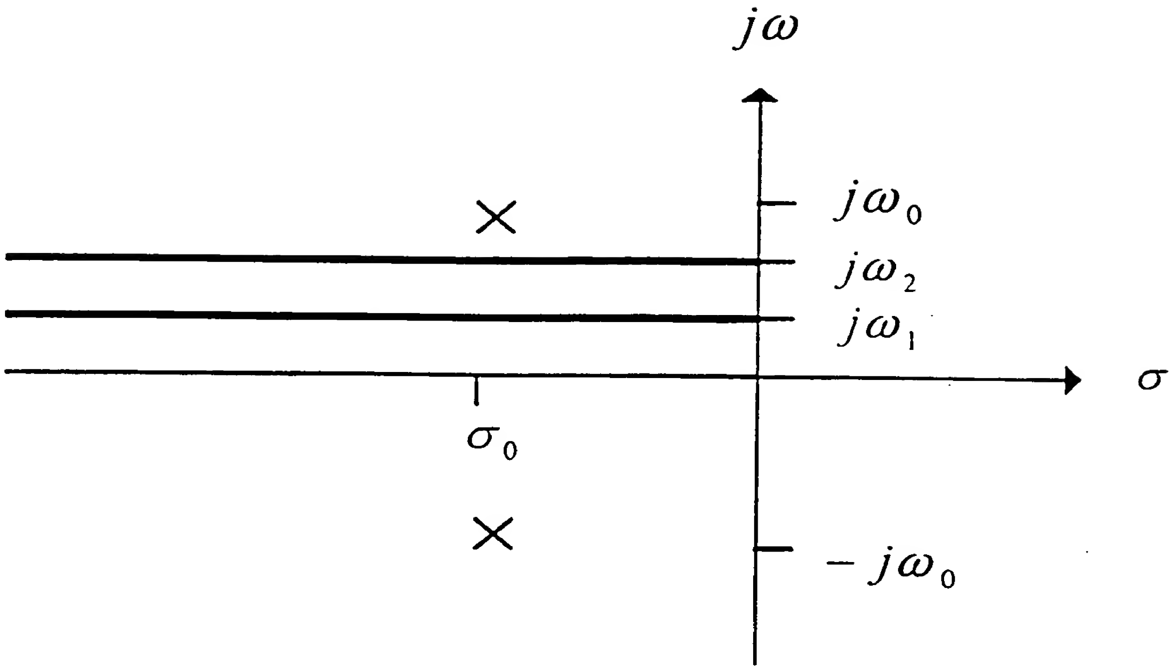

Fig. 4

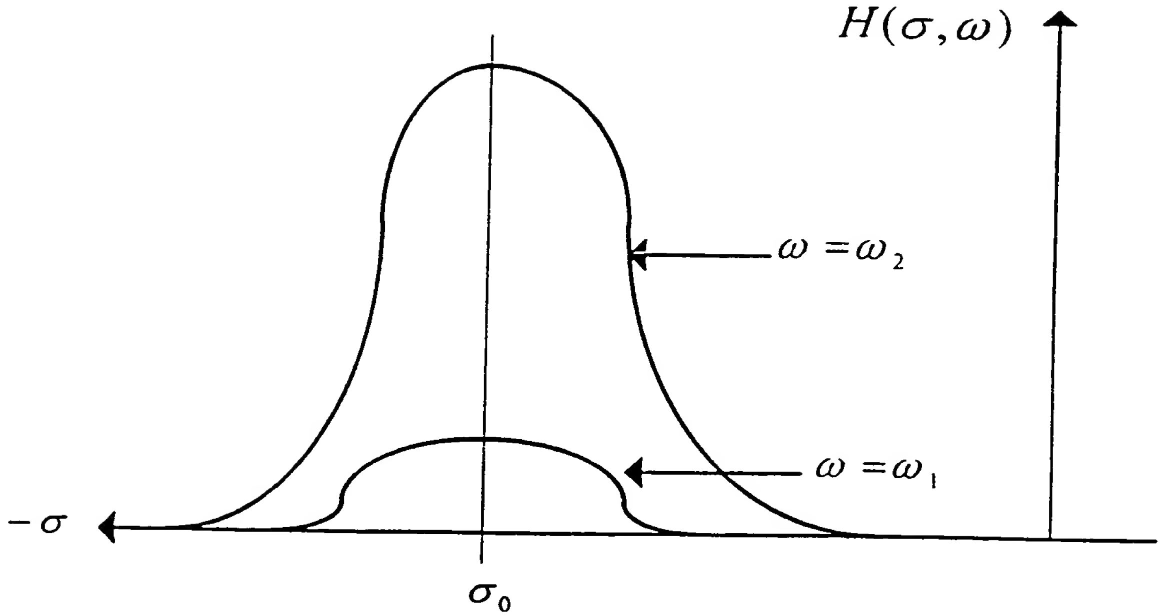

Fig. 5

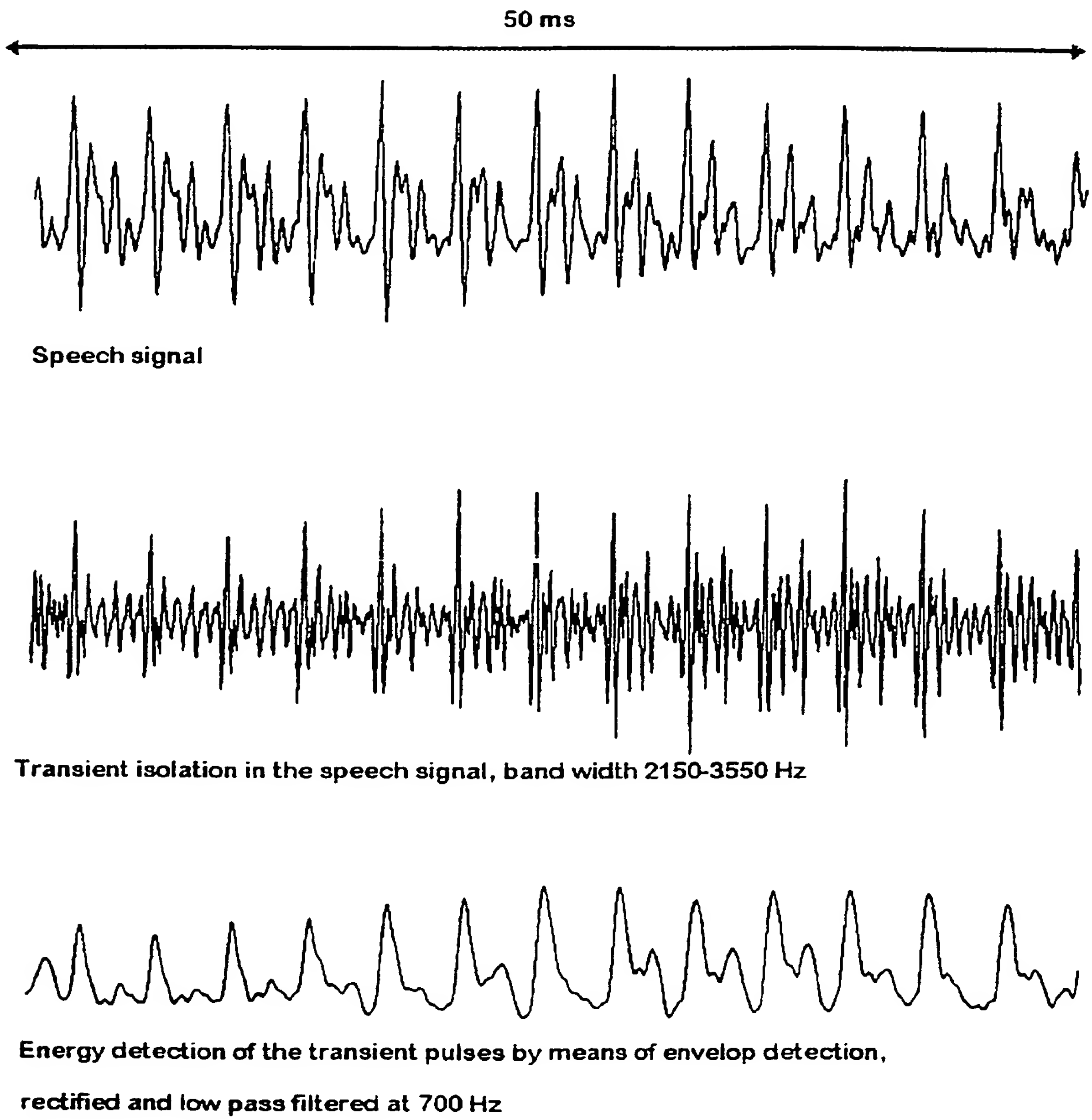
50 ms
Speech signal
Transient isolation in the speech signal, band width 2150-3550 Hz
Energy detection of the transient pulses by means of envelop detection,
rectified and low pass filtered at 700 Hz

Fig. 6

Sigma	Max	
9250	0.931	0.81633
8500	0.964	0.81633
7750	0.989	0.81633
7000	1.000	0.81633
6250	0.993	0.81633
5500	0.960	0.81633
4750	0.895	0.81633
4000	0.855	0.90703
3250	0.757	0.90703
2500	0.610	0.99773

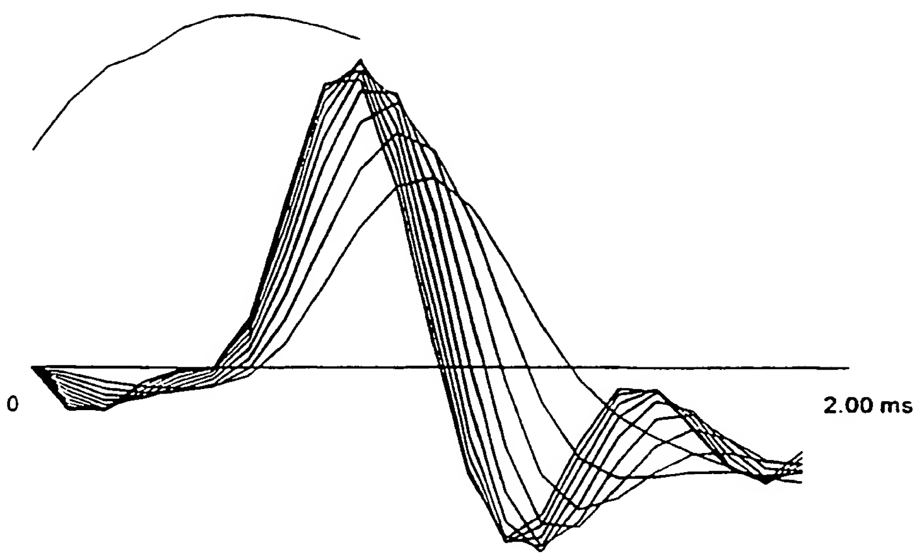

Fig. 7

Sigma	Max	
9250	0.980	0.72562
8500	0.989	0.72562
7750	0.983	0.72562
7000	0.986	0.81633
6250	1.000	0.81633
5500	0.983	0.81633
4750	0.923	0.81633
4000	0.837	0.90703
3250	0.745	0.90703
2500	0.590	0.99773

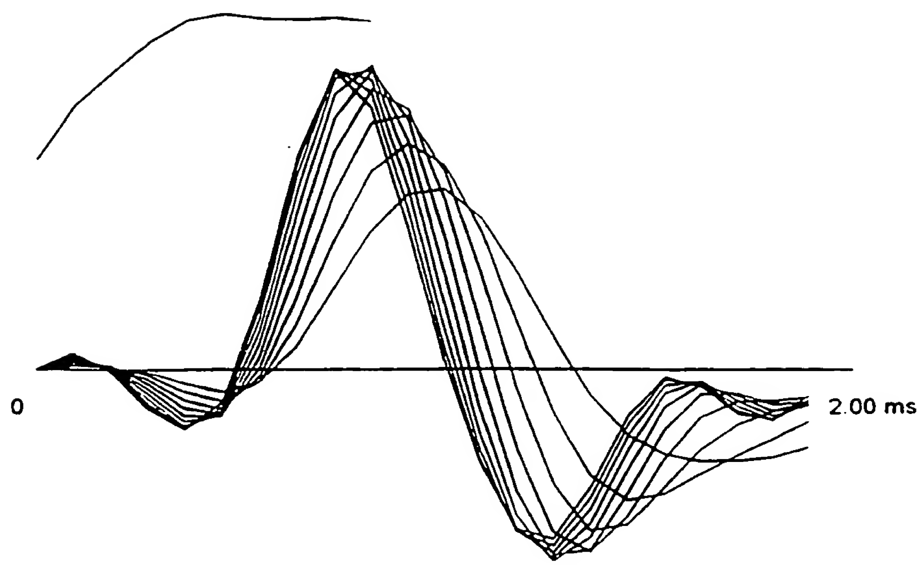

Fig. 8

Sigma	Max	
9250	0.883	0.81633
8500	0.908	0.81633
7750	0.931	0.81633
7000	0.953	0.81633
6250	0.974	0.81633
5500	0.992	0.81633
4750	1.000	0.81633
4000	0.984	0.81633
3250	0.940	0.90703
2500	0.851	0.90703

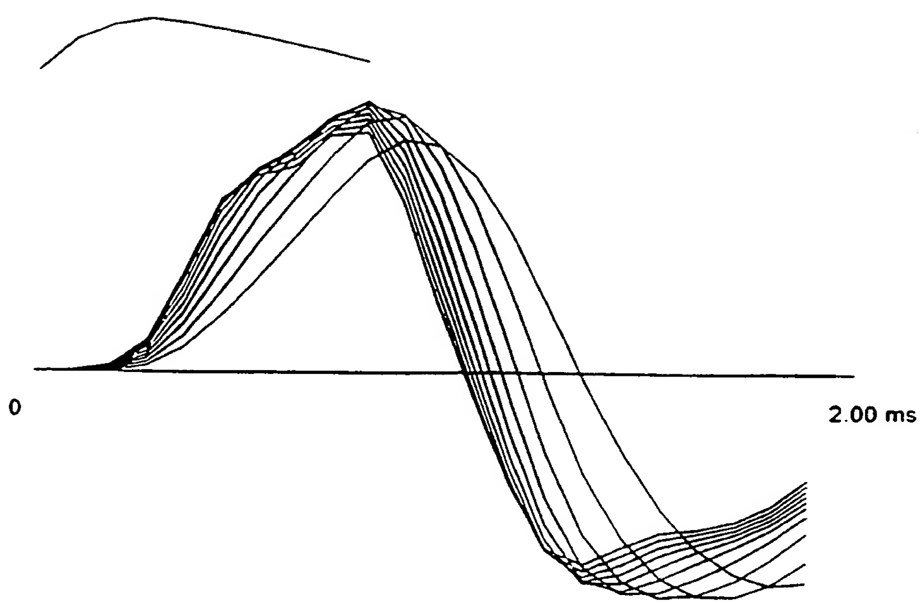

Fig. 9

Sigma	Max	
9250	0.890	0.54422
8500	0.917	0.54422
7750	0.944	0.54422
7000	0.971	0.54422
6250	0.992	0.54422
5500	1.000	0.54422
4750	0.982	0.54422
4000	0.977	0.63492
3250	0.912	0.63492
2500	0.795	0.72562

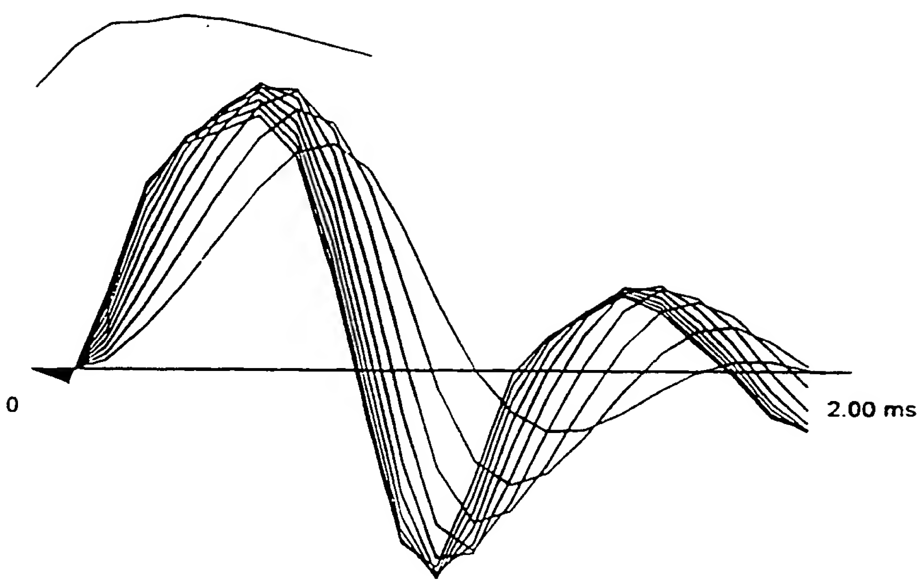

Fig. 10

Sigma	Max	
9250	0.965	0.99773
8500	0.984	0.99773
7750	0.995	0.99773
7000	1.000	0.99773
6250	0.998	0.99773
5500	0.989	0.99773
4750	0.968	0.99773
4000	0.964	1.08844
3250	0.920	1.08844
2500	0.831	1.17914

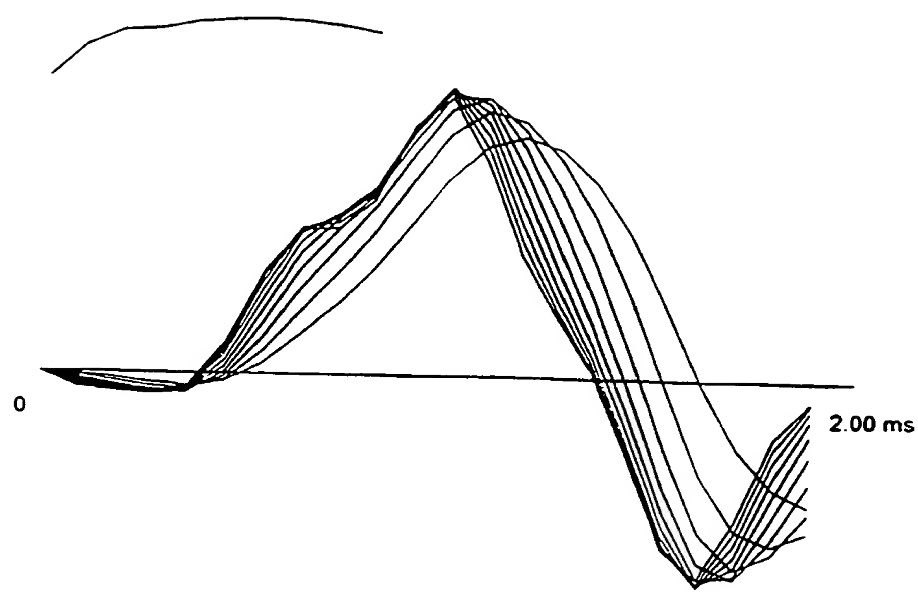

Fig. 11

Sigma	Max	
9250	0.983	0.81633
8500	0.994	0.81633
7750	0.995	0.81633
7000	0.986	0.81633
6250	0.994	0.90703
5500	1.000	0.90703
4750	0.989	0.90703
4000	0.953	0.99773
3250	0.922	0.99773
2500	0.859	1.08844

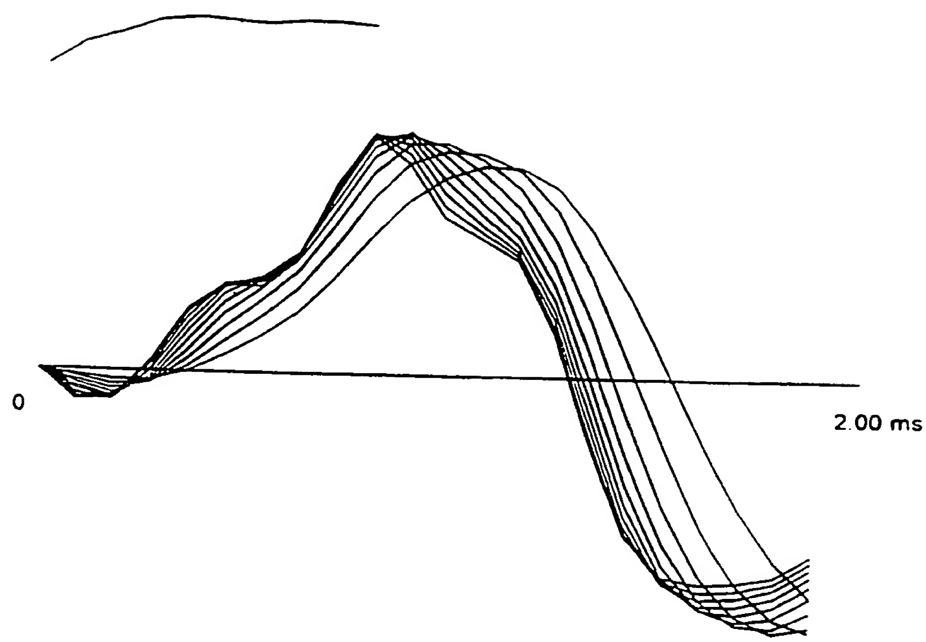

Fig. 12

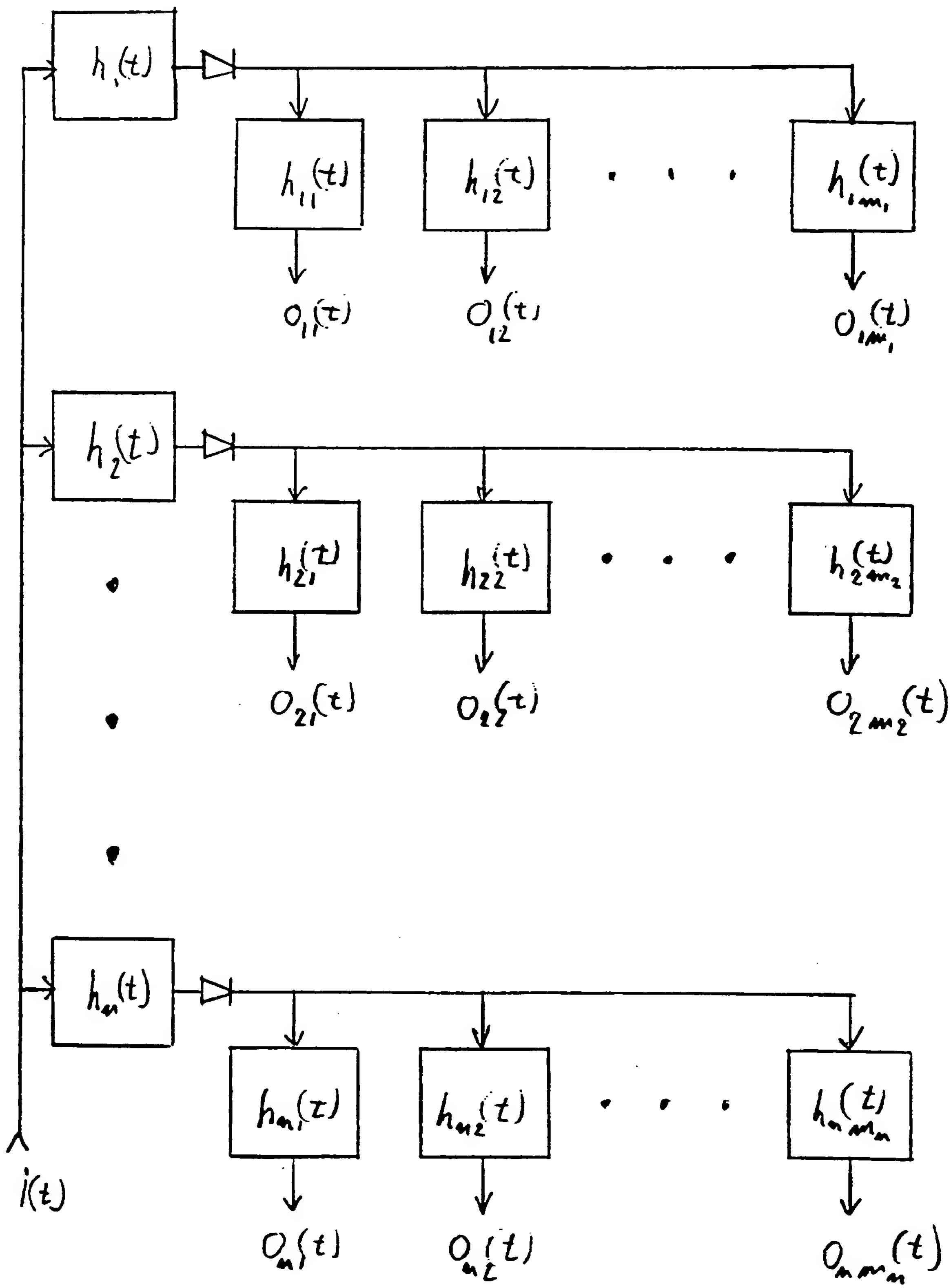
$h_1(t)$
$h_{11}(t)$
$h_{12}(t)$
$h_{1m_1}(t)$
$O_{11}(t)$
$O_{12}(t)$
$O_{1m_1}(t)$
$h_2(t)$
$h_{21}(t)$
$h_{22}(t)$
$h_{2m_2}(t)$
$O_{21}(t)$
$O_{22}(t)$
$O_{2m_2}(t)$
$h_n(t)$
$h_{n1}(t)$
$h_{n2}(t)$
$h_{nm_n}(t)$
$O_{n1}(t)$
$O_{n2}(t)$
$O_{nm_n}(t)$
$i(t)$

Fig. 13

ATENT COOPERATION TR Y

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION OF THE RECORDING OF A CHANGE

(PCT Rule 92bis.1 and Administrative Instructions, Section 422)

To:

HOFMAN-BANG, BOUTARD, LEMAN & REE
A/S
Hans Bekkevolds Allé 7
DK-2900 Hellerup
DANEMARK

Date of mailing (day/month/year) 08 June 2000 (08.06.00)	
Applicant's or agent's file reference 20315 PC 1	**IMPORTANT NOTIFICATION**
International application No. PCT/DK99/00128	**International filing date** (day/month/year) 12 March 1999 (12.03.99)

1. The following indications appeared on record concerning:

☐ the applicant ☐ the inventor ☒ the agent ☐ the common representative

Name and Address	State of Nationality	State of Residence
PLOUGMANN, VINGTOFT & PARTNERS A/S Sankt Annæ Plads 11 P.O. Box 3007 DK-1021 Copenhagen Denmark	Telephone No. Facsimile No. Teleprinter No.	

2. The International Bureau hereby notifies the applicant that the following change has been recorded concerning:

☐ the person ☒ the name ☒ the address ☐ the nationality ☐ the residence

Name and Address	State of Nationality	State of Residence
HOFMAN-BANG, BOUTARD, LEMAN & REE A/S Hans Bekkevolds Allé 7 DK-2900 Hellerup Denmark	Telephone No. 45 39 48 80 00 Facsimile No. 45 39 48 80 80 Teleprinter No.	

3. Further observations, if necessary:
Please note that the agent in box 1 has renounced his appointment as agent of record. All future correspondence should be sent to the new agent in box 2.

4. A copy of this notification has been sent to:

☒ the receiving Office
☐ the International Searching Authority
☐ the International Preliminary Examining Authority
☐ the designated Offices concerned
☒ the elected Offices concerned
☒ other: PLOUGMANN, VINGTOFT & PARTNERS

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland** Facsimile No.: (41-22) 740.14.35	Authorized officer A. Karkachi Telephone No.: (41-22) 338.83.38

'ATENT COOPERATION T TY

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION OF THE RECORDING OF A CHANGE

(PCT Rule 92bis.1 and Administrative Instructions, Section 422)

To:

LEONHARD, Frank, Uldall
Louisevej 13
DK-2800 Lyngby
DANEMARK

Date of mailing (day/month/year) 08 June 2000 (08.06.00)	
Applicant's or agent's file reference 20315 PC 1	**IMPORTANT NOTIFICATION**
International application No. PCT/DK99/00128	**International filing date** (day/month/year) 12 March 1999 (12.03.99)

1. The following indications appeared on record concerning:

☐ the applicant ☐ the inventor ☒ the agent ☐ the common representative

Name and Address	State of Nationality	State of Residence
PLOUGMANN, VINGTOFT & PARTNERS A/S Sankt Annæ Plads 11 P.O. Box 3007 DK-1021 Copenhagen Denmark		
	Telephone No.	
	Facsimile No.	
	Teleprinter No.	

2. The International Bureau hereby notifies the applicant that the following change has been recorded concerning:

☐ the person ☐ the name ☐ the address ☐ the nationality ☐ the residence

Name and Address	State of Nationality	State of Residence
	Telephone No.	
	Facsimile No.	
	Teleprinter No.	

3. Further observations, if necessary:
Please note that the agent in Box 1 has renounced his appointment as agent of record. Please send all future correspondence to the address indicated in the addressee box of this notification.

4. A copy of this notification has been sent to:

☒ the receiving Office
☐ the International Searching Authority
☐ the International Preliminary Examining Authority
☐ the designated Offices concerned
☒ the elected Offices concerned
☒ other: PLOUGMANN, VINGTOFT & PARTNERS

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland** Facsimile No.: (41-22) 740.14.35	Authorized officer A. Karkachi Telephone No.: (41-22) 338.83.38

PATENT COOPERATION TREATY

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION OF ELECTION

(PCT Rule 61.2)

To:

Assistant Commissioner for Patents
United States Patent and Trademark Office
Box PCT
Washington, D.C.20231
ÉTATS-UNIS D'AMÉRIQUE

in its capacity as elected Office

Date of mailing (day/month/year) 27 October 1999 (27.10.99)	
International application No. PCT/DK99/00128	**Applicant's or agent's file reference** 20315 PC 1
International filing date (day/month/year) 12 March 1999 (12.03.99)	**Priority date** (day/month/year) 13 March 1998 (13.03.98)
Applicant LEONHARD, Frank, Uldall	

1. The designated Office is hereby notified of its election made:

   [X] in the demand filed with the International Preliminary Examining Authority on:

   27 September 1999 (27.09.99)

   [ ] in a notice effecting later election filed with the International Bureau on:

2. The election [X] was [ ] was not

   made before the expiration of 19 months from the priority date or, where Rule 32 applies, within the time limit under Rule 32.2(b).

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland** Facsimile No.: (41-22) 740.14.35	Authorized officer F. Baechler Telephone No.: (41-22) 338.83.38

M.H

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PATENT COOPERATION TREATY

PCT

REC'D 28 DEC 1999

WIPO PCT

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

Applicant's or agent's file reference 20315 PC 1	**FOR FURTHER ACTION**	See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)
International application No. PCT/DK99/00128	International filing date *(day/month/year)* ~~15~~03/1999	Priority date *(day/month/year)* 13/03/1998
International Patent Classification (IPC) or national classification and IPC G10L3/02		
Applicant LEONHARD, Frank, Uldall		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 5 sheets, including this cover sheet.

   ☐ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of sheets.

3. This report contains indications relating to the following items:

   I ☒ Basis of the report
   II ☐ Priority
   III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   IV ☐ Lack of unity of invention
   V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations suporting such statement
   VI ☐ Certain documents cited
   VII ☒ Certain defects in the international application
   VIII ☐ Certain observations on the international application

Date of submission of the demand 27/09/1999	Date of completion of this report 22.12.1999
Name and mailing address of the international preliminary examining authority: European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Authorized officer La Gioia, C Telephone No. +49 89 2399 2418

Form PCT/IPEA/409 (cover sheet) (January 1994)

I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

Description, pages:

1-25 as originally filed

Claims, No.:

1-22 as originally filed

Drawings, sheets:

1/13-13/13 as originally filed

2. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:
- ☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. Statement

Novelty (N)	Yes:	Claims	1-22
	No:	Claims	
Inventive step (IS)	Yes:	Claims	1-22
	No:	Claims	
Industrial applicability (IA)	Yes:	Claims	1-22
	No:	Claims	

2. Citations and explanations

see separate sheet

VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

see separate sheet

SECTION V

A. The following documents have been considered for the purposes of this report:

D1: HALIJAK C A ET AL: 'Simple consequences of the finite time Laplace ~~transform analysis~~ ~~of~~ the periodically reversed switched capacitors' CIRCUITS, SYSTEMS, AND SIGNAL PROCESSING, 1985, USA, vol. 4, no. 4, pages 503-511, XP002105446 ISSN 0278-081X

D2: BARRETT T W: 'The cochlea as Laplace analyzer for optimum (elementary) signals', ACUSTICA, FEB. 1978, WEST GERMANY, vol. 39, no. 3, pages 155-172, XP002105445 ISSN 0001-7884

D3: HARBOR R D ET AL: 'THE LAPLACE TRANSFORM', ENERGY AND INFORMATION TECHNOLOGIES IN THE SOUTHEAST, COLUMBIA, APRIL 9 - 12, 1989, vol. 1, pages 376-379, XP000076824, IEEE

B. The present application satisfies the criteria set forth in Article 33(1)-(3) PCT because the subject-matter of independent claims 1 and 19 is novel in respect of the presently available prior art and involves an inventive step (Rule 65(1)(2) PCT), since the presently available prior art neither discloses nor renders obvious determining a parameter of a system generating a signal for further classification of the system or for further signal processing by executing a short time Laplace transformation of the signal in accordance with the formula set out in the independent claims.

B.1 The dependent claims add further features to the respective independent claims and thus also relate to novel and inventive subject-matter.

SECTION VII

A. The documents D1, D2 and D3, offering relevant background art as regards the Laplace transform, should have been identified in the description; Rule 5.1(a)(ii)

PCT.

ATENT COOPERATION TREAT

PCT

INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

Applicant's or agent's file reference	FOR FURTHER ACTION	see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below.
20315 PC 1		
International application No. PCT/DK 99/ 00128	International filing date (day/month/year) 12/03/1999	(Earliest) Priority Date (day/month/year) 13/03/1998
Applicant LEONHARD, Frank, Uldall		

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 3 sheets.

[X] It is also accompanied by a copy of each prior art document cited in this report.

1. **Basis of the report**

   a. With regard to the **language**, the international search was carried out on the basis of the international application in the language in which it was filed, unless otherwise indicated under this item.

      [ ] the international search was carried out on the basis of a translation of the international application furnished to this Authority (Rule 23.1(b)).

   b. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international search was carried out on the basis of the sequence listing :

      [ ] contained in the international application in written form.
      [ ] filed together with the international application in computer readable form.
      [ ] furnished subsequently to this Authority in written form.
      [ ] furnished subsequently to this Authority in computer readble form.
      [ ] the statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.
      [ ] the statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished

2. [ ] **Certain claims were found unsearchable** (See Box I).

3. [ ] **Unity of invention is lacking** (see Box II).

4. With regard to the **title,**

   [ ] the text is approved as submitted by the applicant.
   [X] the text has been established by this Authority to read as follows:

   A SIGNAL PROCESSING METHOD TO ANALYSE TRANSIENTS OF SPEECH SIGNALS

5. With regard to the **abstract,**

   [X] the text is approved as submitted by the applicant.
   [ ] the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this international search report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is Figure No. 13

   [ ] as suggested by the applicant.
   [X] because the applicant failed to suggest a figure.
   [ ] because this figure better characterizes the invention.

   [ ] None of the figures.

INTERNATIONAL SEARCH REPORT

International Application No
PCT/DK 99/00128

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 G10L3/02 G10L5/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	BARRETT T W: "The cochlea as Laplace analyzer for optimum (elementary) signals" ACUSTICA, FEB. 1978, WEST GERMANY, vol. 39, no. 3, pages 155-172, XP002105445 ISSN 0001-7884 see page 159, column 2, line 10 - line 31 ---	1,7,11, 12,19
A	HARBOR R D ET AL: "THE LAPLACE TRANSFORM" ENERGY AND INFORMATION TECHNOLOGIES IN THE SOUTHEAST, COLUMBIA, APRIL 9 - 12, 1989, vol. 1, 9 April 1989, pages 376-379, XP000076824 INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS * Equation (1a) * --- -/--	1,7,9, 11,12,19

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
9 June 1999	23/06/1999

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Krembel, L

1

Form PCT/ISA/210 (second sheet) (July 1992)

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
A	HALIJAK C A ET AL: "Simple consequences of the finite time Laplace transform analysis of the periodically reversed switched capacitors" CIRCUITS, SYSTEMS, AND SIGNAL PROCESSING, 1985, USA, vol. 4, no. 4, pages 503-515, XP002105446 ISSN 0278-081X * Equation (1) * ---	1,7,11, 12,19
A	CELEBI S ET AL: "Analysis of spectral feature extraction using the gamma filter" 1994 IEEE INTERNATIONAL CONFERENCE ON NEURAL NETWORKS. IEEE WORLD CONGRESS ON COMPUTATIONAL INTELLIGENCE (CAT. NO.94CH3429-8), PROCEEDINGS OF 1994 IEEE INTERNATIONAL CONFERENCE ON NEURAL NETWORKS (ICNN'94), ORLANDO, FL, USA, 27 JUNE-2 JULY 1994, pages 4497-4501 vol.7, XP002105447 ISBN 0-7803-1901-X, 1994, New York, NY, USA, IEEE, USA * Paragraph "Gamma Network" * ---	1,7,11, 12,19
A	WO 94 25958 A (LEONHARD FRANK ULDALL) 10 November 1994 cited in the application see abstract -----	1,7,9, 11,12,19

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/DK 99/00128

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9425958 A	10-11-1994	AT 178155 T	15-04-1999
		AU 6535994 A	21-11-1994
		CN 1125010 A	19-06-1996
		DE 69417445 D	29-04-1999
		EP 0737351 A	16-10-1996
		FI 955025 A	15-12-1995
		JP 8509556 T	08-10-1996
		US 5884260 A	16-03-1999